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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/799,533	03/11/2004	Yang Gao	0160112	8500

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FARJAMI & FARJAMI LLP
26522 LA ALAMEDA AVE.
SUITE 360
MISSION VIEJO, CA 92691

EXAMINER	
JACKSON, JAKIEDA R	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/799,533	GAO, YANG	
	Examiner	Art Unit	
	Jakieda R. Jackson	2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-52 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-52 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on 24 September 2003 is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. **Claims 1-5, 7-21, 23-38 and 40-45** are rejected under 35 U.S.C. 103(a) as being unpatentable over Bergstrom et al. (USPN 5,809,459), hereinafter referenced as Bergstrom in view of Kaajas et al. (PGPUB 2004/0138874), hereinafter referenced as Kaajas.

Regarding **claims 1 and 17**, Bergstom discloses a method and apparatus of processing speech comprising:

obtaining an input speech signal (input speech; column 3, line 63 – column 4, line 24);

processing said voiced portion of said input speech to obtain a first set of parameters using analysis by synthesis approach (analysis and synthesis processor; column 3, line 63 – column 4, line 24); and

processing said noise portion of said input speech to obtain a second set of parameters using open loop approach (open loop; column 12, lines 5-22), but does not specifically teach decomposing said input speech into a voiced portion and a noise portion using an adaptive separation component.

Kaajas teaches audio signal processing using CELP comprising decomposing said input speech into a voiced portion and a noise portion using an adaptive separation component (separate voiced/unvoiced; columns 1-2, paragraphs 0019-0022), to increase the coding gain.

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Bergstom's method and apparatus wherein it decomposes said input speech into a voiced portion and a noise portion using an adaptive separation component, as taught by Kaajas, to increase the coding gain which enhances spatial processing (column 1, paragraphs 0012-0013).

Regarding **claims 2 and 18**, Bergstrom in view of Kaajas disclose everything as claimed in claims 1 and 17. In addition Kaajas discloses a method and apparatus wherein said input speech signal excludes background noise (columns 1-2, paragraphs 0019-0022).

Regarding **claims 3 and 19**, Bergstrom discloses a method and apparatus wherein said separation component is a lowpass filter (lowpass filter; column 6, lines 29-66).

Regarding **claims 4 and 20**, Bergstrom discloses a method and apparatus, wherein bandwidth of said lowpass filter is dependent upon a characteristic of said input speech (lowpass filter; column 6, lines 29-66).

Regarding **claims 5 and 21**, Bergstrom discloses a method and apparatus wherein said characteristic of said input speech is pitch correlation (pitch; column 6, lines 29-66).

Regarding **claims 7 and 23**, Bergstrom in view of Kaajas disclose everything as claimed in claims 1 and 17. In addition Kaajas discloses a method and apparatus wherein said analysis by synthesis approach is a Code Excited Linear Prediction (CELP) process (columns 1—2, paragraphs 0019-0022)..

Regarding **claims 8, 24, 34 and 41**, Bergstrom discloses a method and apparatus wherein said first set of parameters comprises pitch of said voiced portion of said input speech (pitch; column 6, lines 29-66).

Regarding **claims 9, 25, 35 and 42**, Bergstrom discloses a method and apparatus wherein said first set of parameters comprises excitation of said voiced portion of said input speech (excitation; column 6, lines 29-66 and column 8, lines 19-26).

Regarding **claims 10, 26, 36 and 43**, Bergstrom discloses a method and apparatus wherein said first set of parameters comprises energy of said voiced portion of said input speech (energy; column 8, lines 19-26).

Regarding **claims 11, 27, 37 and 44**, Bergstrom discloses a method and apparatus wherein said second set of parameters comprises characteristics of a voicing index of said input speech (index; column 6, lines 29-66 and column 7, lines 22-53).

Regarding **claims 12 and 28**, Bergstrom discloses a method and apparatus further comprising:

transmitting information regarding said first set of parameters to said decoder device (decoding device; column 4, lines 14-24).

Regarding **claims 13 and 29**, Bergstrom discloses a method and apparatus wherein said decoder device uses said information regarding said first set of parameters to synthesize said voiced portion of said input speech (synthesized speech; column 4, lines 6-13).

Regarding **claims 14 and 30**, Bergstrom discloses a method and apparatus further comprising:

transmitting information regarding said second set of parameters to said decoder device (decoding device; column 4, lines 14-24).

Regarding **claims 15 and 31**, Bergstrom discloses a method and apparatus wherein said decoder device uses said information regarding said second set of parameters to synthesize said noise portion of said input speech (synthesized speech; column 4, lines 6-13).

Regarding **claims 16 and 32**, Bergstrom discloses a method and apparatus further comprising:

transmitting a voicing index to said decoder device for synthesizing said input speech (synthesized speech; column 4, lines 6-13 with column 6, lines 29-66 and column 7, lines 22-53).

Regarding **claims 33 and 40**, Bergstrom discloses an apparatus and method for synthesizing speech comprising:

a first module for obtaining a first set of parameters regarding a voiced portion of an input speech signal (input speech; column 3, line 63 – column 4, line 24);

a second module for obtaining a second set of parameters regarding a noise portion of said input speech signal (noise; column 20, lines 1-33);

a third module for synthesizing said voiced portion of said input speech signal from said first set of parameters approach (analysis and synthesis processor; column 3, line 63 – column 4, line 24);

a fourth module for synthesizing said noise portion of said input speech signal from said second set of parameters (noise; column 20, lines 1-33; and

a fifth module for combining said synthesized voiced portion and said synthesized noise portion to produce a synthesized version of said input speech (synthesize; column 4, lines 6-24), but does not specifically teach decomposing said input speech into a voiced portion and a noise portion using an adaptive separation component.

Kaajas teaches audio signal processing using CELP comprising decomposing said input speech into a voiced portion and a noise portion using an adaptive separation component (separate voiced/unvoiced; columns 1-2, paragraphs 0019-0022), to increase the coding gain.

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Bergstom's method and apparatus wherein it decomposes said input speech into a voiced portion and a noise portion using an adaptive separation component, as taught by Kaajas, to increase the coding gain which enhances spatial processing (column 1, paragraphs 0012-0013).

Regarding **claims 38 and 45**, it is interpreted and rejected for the same reasons as set forth in claims 1 and 17 above, in addition, Bergstrom discloses an apparatus and method wherein said second set parameters comprises characteristics of a lowpass filter used for separating said voiced portion (voiced) and said noise portion of said input speech at source of said noise portion (unvoiced; columns 1-2, paragraphs 0019-0022).

3. **Claims 6, 22, 39, 46-48 and 50-51** are rejected under 35 U.S.C. 103(a) as being unpatentable over Bergstrom in view of Kaajas and in further view of Kingsbury et al. (USPN 6,308,155), hereinafter referenced as Kingsbury.

Regarding **claims 6 and 22**, Bergstrom in view of Kaajas discloses a method and apparatus for processing speech, but does not specifically teach wherein said characteristic of said input speech is gender of a person uttering said input speech.

Kingsbury teaches a speech processing method and apparatus wherein said characteristic of said input speech is gender of a person uttering said input speech (gender; column 3, lines 30-47), to capture linguistic content.

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Bergstrom in view of Kaajas' method and apparatus wherein said characteristic of said input speech is gender of a person uttering said input speech, as taught by Kingsbury, to optimize and improve the overall performance of the recognizer (column 3, lines 30-47).

Regarding **claims 39 and 46**, Bergstrom in view of Kaajas discloses a method and apparatus for processing speech, but does not specifically wherein said synthesized noise portion is estimated.

Kingsbury teaches a speech processing method and apparatus wherein said synthesized noise portion is estimated (estimate; column 8, lines 39-57), to improve overall performance.

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Bergstrom in view of Kaajas' method and apparatus wherein said synthesized noise portion is estimated, as taught by Kingsbury, to optimize the performance by reducing the error rate of automatic speech recognizers under degraded acoustic conditions (column 2, lines 45-57).

Regarding claims **47 and 50** Bergstrom in view of Kaajas discloses a method and apparatus for processing speech, but does not specifically a method and apparatus further comprising transmitting a voicing index to a decoder, wherein said voicing index provides filter cut-off frequency for signal decomposition.

Kingsbury teaches a speech processing method and apparatus further comprising transmitting a voicing index to a decoder, wherein said voicing index provides filter cut-off frequency for signal decomposition (column 2, lines 3-14), to improve overall performance.

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Bergstrom in view of Kaajas' method and apparatus wherein it further comprises transmitting a voicing index to a decoder,

wherein said voicing index provides filter cut-off frequency for signal decomposition, as taught by Kingsbury, to optimize the performance by reducing the error rate of automatic speech recognizers under degraded acoustic conditions (column 2, lines 45-57).

Regarding claims **48 and 51**, Bergstrom in view of Kaajas discloses a method and apparatus for processing speech, but does not specifically a method and apparatus wherein said filter cut-off frequency is communicated to said decoder using a plurality of bits in said voicing index to indication to said decoder which filter to use for said signal decomposition.

Kingsbury teaches a speech processing method and apparatus wherein said filter cut-off frequency is communicated to said decoder using a plurality of bits in said voicing index to indication to said decoder which filter to use for said signal decomposition (column 2, lines 3-14), to improve overall performance.

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Bergstrom in view of Kaajas' method and apparatus wherein said filter cut-off frequency is communicated to said decoder using a plurality of bits in said voicing index to indication to said decoder which filter to use for said signal decomposition, as taught by Kingsbury, to optimize the performance by reducing the error rate of automatic speech recognizers under degraded acoustic conditions (column 2, lines 45-57).

4. **Claims 49 and 52** are rejected under 35 U.S.C. 103(a) as being unpatentable over Bergstrom in view of Kaajas and Kingsbury, and in further view of Li et al. (PGPUB 2007/0110042), hereinafter referenced as Li.

Regarding claims **49 and 52**, Bergstrom in view of Kaajas and Kingsbury disclose a method and apparatus for processing speech, but does not specifically teach wherein said voicing index defines a plurality of low pass filters.

Li discloses a speech processing method and apparatus wherein said voicing index defines a plurality of low pass filters (column 22, paragraphs 0250-0251 with column 27, paragraphs 0280-0281), to optimize the performance.

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Bergstrom in view of Kaajas and Kingsbury's method and apparatus wherein said voicing index defines a plurality of low pass filters, as taught by Li, to multiply the down-sampled signal by cosine and sine, which optimizes the performance (column 22, paragraphs 0250-0251).

Conclusion

5. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

- Adut (USPN 6,985,857) discloses a method and apparatus for speech coding using training and quantizing.

- Paetzold et al. (USPN 6,940,454) disclose a method and system for generating facial animation values based on a combination of visual and audio information.
- Su et al. (USPN 6,014,622) disclose a low bit rate speech coder using adaptive open-loop subframe pitch lag estimation and vector quantization.
- Chen et al. (USPN 5,884,010) disclose linear prediction coefficient generation during frame erasure or packet loss.

6. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Jakieda R. Jackson whose telephone number is 571-272-7619. The examiner can normally be reached on Monday, Tuesday and Thursday 7:30 a.m. to 5:00p.m.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, David Hudspeth can be reached on 571-272-7843. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

JRJ
June 12, 2007

DAVID HUDSPETH
SUPERVISORY PATENT EXAMINER
TECHNOLOGY CENTER 2600